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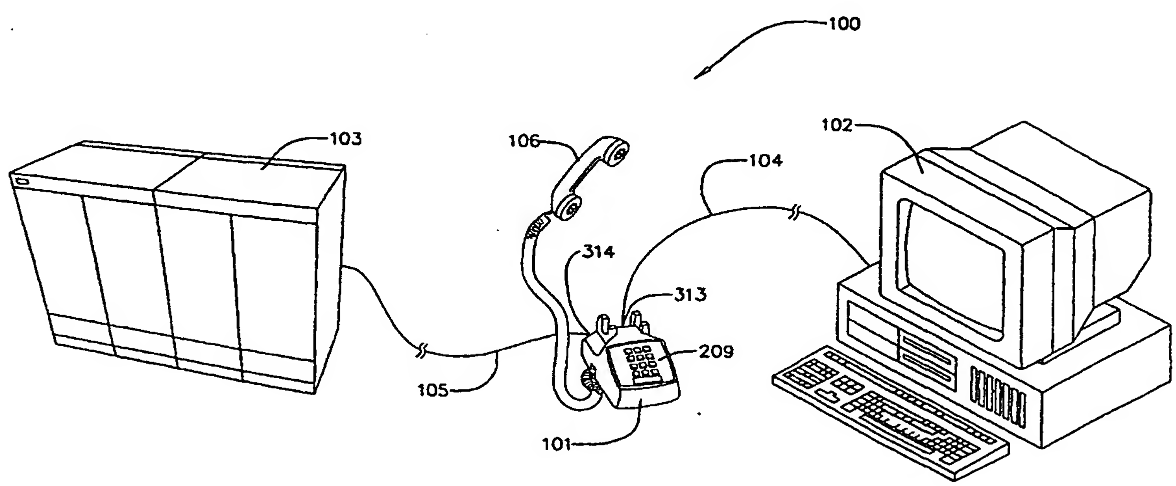

(57) Abstract

The present invention includes a network telephone having a microphone coupled to provide voice data to a network, a speaker coupled to facilitate listening to voice data from the network, a dialing device coupled to facilitate routing of voice data upon the network, a first port configured to facilitate communication with a first network device, a second port configured to facilitate communication with a second network device and a prioritization circuit coupled to apply prioritization to voice data provided by the microphone.

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NETWORK TELEPHONY SYSTEM

RELATED APPLICATION

This patent application claims the benefit of the filing date of United States Provisional Patent Application Serial No. 60/110,211, filed November 30, 1998 and entitled 3 PORT ETHERNET SWITCH AND MICROPROCESSOR FOR VOICE COMMUNICATION, the entire contents of which are hereby expressly incorporated by reference.

FIELD OF THE INVENTION

The present invention relates generally to computer network systems. The present invention relates more particularly to a system for facilitating telephony over a computer network, such as an Ethernet network.

BACKGROUND OF THE INVENTION

Ethernet networks and the like for providing data communication among a plurality of computers are well-known. Such networks facilitate the transfer of data files, audio information and video information, as well as any other information which may be represented in binary form, among the plurality of computers.

Typically, one or more of the computers is configured as a server and generally defines a repository for frequently used files. The other, e.g., non-server, computers are generally referred to as clients and may frequently receive files from the server. Client computers may also communicate information to one another.

Although common, servers are not a necessary part of all networks. In peer-to-peer networks, client or non-server computers communicate among one another to facilitate file transfer.

Networks can be conveniently divided into two broad categories, based upon their size. A local area network (LAN) is a group of computers which are connected so as to facilitate the sharing of applications, data and peripherals. Local area networks are generally confined to a single building or a small group of buildings.

A wide area network (WAN) is made up of a plurality of LANs which are connected together so as to facilitate communication therebetween. A WAN may cover a city, a state, a country or even be international in scope. The Internet is an example of a WAN that includes more than 2,000 separate packet-switched networks that are located all over the world.

Networks, particularly WANs, are typically interconnected by a variety of network devices such as hubs, switches, routers and/or bridges.

A hub is a multiport repeater that facilitates the interconnection of a plurality of computers (one for each port of the hub).

A switch is a network device which is capable of reading and modifying header information associated with data packets, including header information which specifies the priority with which the data packets are to be queued within a buffer of a network device (including the switch itself).

A router is a network device that interconnects a plurality of separate LANs or WANs, wherein each of the networks utilizes the same network protocol and operates at the network layer, or Layer 3, of the ISO model.

A bridge is a network device that interconnects a plurality of separate LANs or WANs, wherein at least two of the networks utilize a different network protocol with respect to one another and operates at the Data Link/MAC layer, or Layer 2 of the ISO model.

The popularity of the Internet has increased the desire for additional network services such as network telephony. The vast, high bandwidth network which defines the Internet provides an ideal medium for audio communications.

Thus, it is desirable to provide a system for facilitating audio communication over networks such as the Internet.

SUMMARY OF THE INVENTION

The present invention includes a network telephone having a microphone coupled to facilitate provision of voice data to a network, a speaker coupled to facilitate listening to voice data from the network, a dialing device coupled to facilitate routing of voice data upon the network, a first port configured to facilitate communication with a first network device, a second port configured to facilitate communication with a second network device and a prioritization circuit coupled to apply prioritization to voice data provided by the microphone.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a drawing showing the network telephone of the present invention electrically installed between a first network device, e.g., a personal computer and a second network device, e.g., a network switch;

FIG. 2 is a block diagram showing the network telephone generally;

FIG. 3 is a block diagram showing the Internet Protocol switch controller of the present invention;

FIG. 4 is a block diagram showing the voice engine processor of the present invention;

FIG. 5 is a flowchart showing processing for an incoming voice packet;

FIG. 6 is a flowchart showing processing for an outgoing voice data packet; and

FIG. 7 is a flowchart showing differences in processing for low priority (non-voice data) and high priority (voice data) packets.

DETAILED DESCRIPTION OF THE INVENTION

The detailed description set forth below in connection with the appended drawings is intended as a description of the presently preferred embodiment of the invention and is not intended to represent the only form in which the present invention may be constructed or utilized. The description sets forth the functions and the sequence of steps for constructing and operating the invention in connection with the illustrated embodiment. It is to be understood, however, that the same or equivalent functions and sequences may be accomplished by different embodiments that are also intended to be encompassed within the spirit and scope of the invention.

Although the present invention is described below and illustrated in the drawings as being configured for use in an Ethernet network, those skilled in the art will appreciate that the network telephone of the present invention is likewise suitable for use in various other network environments. Thus, description and illustration of the network telephone in an Ethernet network is by way of example only and not by way of limitation.

The present invention includes a network telephone having a microphone coupled to facilitate provision of voice data to a network, a speaker coupled to facilitate listening to voice data from the network, a dialing device coupled to facilitate routing of voice data upon the network, a first port configured to facilitate communication with a first network device, a second port configured to facilitate communication with a second network device and a prioritization circuit coupled to apply prioritization to voice data provided by the microphone and detect prioritization of traffic received from either one or both of the network ports. The prioritization circuit is optionally coupled to apply prioritization to voice data communicated to the network telephone via the first and/or second ports.

The first port is configured to facilitate communication of voice packets and non-voice packets with the first network device such as a personal computer, and the second port is similarly configured to facilitate communication of voice packets and non-voice packets with the second network device such as a network switch. The first and second ports optionally comprise Ethernet 10/100 ports.

The microphone and the speaker are optionally part of a handset. The dialing device may be defined by a keypad. However, those skilled in the art will appreciate that various other types of dialing devices, e.g., touchpads, voice control, etc., are likewise suitable.

According to one aspect of the present invention, the prioritization circuit is configured to tag voice packets, so as to facilitate prioritization thereof. The prioritization circuit is also configured to read tags associated with data packets provided to the network telephone by the network. Thus, data packets being transmitted from the network telephone to the network are processed by the network according to the priority assigned by the prioritization circuit and data packets received by the network telephone are similarly processed according to their assigned priority, as discussed in detail below.

According to [illegible]spect of the present invention, the pr[illegible]zation is defined by a network switch, such as an Ethernet switch. As those skilled in the art will appreciate, a network switch may be configured to add tags to data packets which pass therethrough, so as to associate a priority with the data packets. The priority determines the order of processing by various network devices. The integral network switch of the network telephone of the present invention functions as a network switch controller, since the integral network switch of the network telephone uses such tags to effect control of network switches throughout the network.

A voice engine processor in communication with the network switch is configured to digitize and compress voice data from the microphone and to decompress and perform digital-to-analog conversion upon voice data provided to the speaker. The voice engine processor is also configured to depacketize voice data which is being provided to the speaker (typically from the network) and to packetize voice data which is provided by the microphone.

Thus, according to the present invention, the Internet Protocol switch controller functions as a network switch, so as to facilitate the prioritization of voice data packets. As those skilled in the art will appreciate, the prioritization of voice data packets tends to assure that the voice data packets are not undesirably delayed as they are routed across a network, such as the Internet, from the source network telephone to another network telephone or the like.

Of course, such delays are undesirable because they cause the speech to be interrupted or broken, and may, indeed, result in lost voice packets which cause the speech to be garbled or otherwise unintelligible.

By adding prioritization to voice data which originates at the network telephone of the present invention, the voice data packets tends to be queued ahead of other, lower priority, data packets at various network devices which perform buffering, such as switches, routers, bridges, hubs and the like. The addition of priority to voice data packets assures prompt processing of the voice data packets by network devices.

The network telephone of the present invention is suitable for use in various different types of networks, including but not limited to LANs, WANs, intranets and internets, as well as the Internet.

Referring now to FIG. 1, a network telephone system 100 includes a network telephone 101 electrically connected between a first network device such as a personal computer 102 and a second network device, such as a hub, router, bridge, other network computer or switch 103. The network telephone 101 is electrically connected to the personal computer 102 via cable 104 which attaches to a first input/output port 313 of the network telephone 101. Similarly, the Internet telephone 101 is connected to the network switch 103 via cable 105 which attaches to input/output port 314 of the network telephone 101.

Thus, according to the present invention, installation of a network telephone 101 simply involves disconnecting an existing cable between a personal computer 102 and a network device such as a network switch 103, and connecting the network telephone 100 between the personal

computer 102 and the network switch 103. Thus, the network telephone 101 is merely inserted in series between the personal computer 102 and the network switch 103 such that the network telephone may then intercept voice data packets from the network and may insert voice data packets onto the network.

Not only must the Internet telephone 101 facilitate transmission and reception of voice data packets to and from the network but the network telephone 101 must also allow data packets from the network to be transmitted to the personal computer 102 and allow data packets transmitted from the personal computer 102 to be forwarded to the network.

However, as those skilled in the art will appreciate, the personal computer 102 is not required for the network telephone 101 to function. Thus, the network telephone 101 may alternatively be electrically connected only to the network switch 103 or some other network device. Indeed, those skilled in the art will appreciate that the network telephone 101 may be connected in a variety of different manners to the network, including but not limited to a radio connection, an infrared connection, a fiber optic connection or any other desired type of connection.

Although FIG. 1 depicts a conventional telephone, having a keypad 204 and a handset 106, those skilled in the art will appreciate that various other telephony devices are likewise suitable. For example, the network telephone 101 may comprise a headset and may alternatively utilize voice recognition for dialing. Indeed, the network telephone 101 may define any desired configuration of voice telephone.

Referring now to FIG. 2, the network telephone 101 comprises an Internet Protocol switch controller 201 and a voice engine processor 202 which cooperate to facilitate telephonic communication via a network, such as the Internet. The Internet Protocol switch 201 and the voice engine processor 202 are each formed as separate, single chips. However, future integration may allow these as well as other functions to be implemented in a single chip. The Internet Protocol switch controller 201 facilitates the application of enhanced priority to voice packets in order to assure that the voice packets are not undesirably delayed during transmission to a recipient via the network, as described in detail below. The voice engine processor 202 performs analog-to-digital conversion upon voice from microphone 211, compresses the digitized voice and packetizes the digitized voice for transmission upon the network. The voice engine processor 211 also depacketizes, decompresses and performs digital-to-analog conversion upon voice information from the network such that the voice information may be listened via speaker 212, as also described in detail below

The Internet Protocol switch controller 201 substantially comprises a network switch which is capable of adding tags to network data packets, as well as reading existing tags of network packets. Priority is applied to the network packets via the use of such tags and existing priority information is read from network packets having such tags.

The Internet Protocol switch controller 201 may be configured to accommodate maximum frame sizes of both 1522 and 1548 bytes.

1522 bytes maximum frame length supports the IEEE 802.3ac VLAN tag specification while 1548 bytes allow alternate tagging schemes, with up to 30 bytes of tag field (i.e., user's ISL). Generally, packetized voice will consist of minimax data frames generally of 64 bytes, or 68 bytes with the 802.3ac tag.

The Internet Protocol switch controller 201 interfaces with the voice engine processor 202 via a data interface, such as media independent interface 203 and with a control interface, such as microprocessor interface 204. Those skilled in the art will appreciate that various other types of interfaces for facilitating communication between the Internet Protocol switch controller 201 and the voice engine processor 202 are likewise suitable.

Magnetics 207 and 208 provide DC isolation and impedance matching at media dependent interfaces 313 and 314, respectively. Media dependent interface 313 facilitates connection to the personal computer 102 (FIG. 1) or the like via network interface adapter (NIC) connection 205 and media dependent interface 314 similarly provides communication to a network, e.g., network switch 103 via switch connector 206.

The voice engine processor 202 is in communication with keypad 209, which is used to facilitate desired routing of the voice packets from the network telephone 101, through the network, to a desired recipient. It is worthwhile to note that the desired recipient may be a similar Internet telephone, a computer such as a personal computer 102, a server which archives the voice packets for later use, or a conventional telephone accessed via the public switched telephone network. The voice data provided by the network telephone 101 may leave the network and be transmitted via the public switch telephone system.

Indeed, a voice message provided by the network telephone 101 may, if desired, be converted into a text message via voice recognition software. The text message may then be routed to a computer, such as via e-mail, may be routed to a fax machine, may be routed to an alpha/numeric pager or may be routed to any other desired text device.

The Internet connection between the network telephone 101 and the personal computer 102 preferably includes an Ethernet 10/100 connection. Similarly, the connection between the Internet telephone 101 and the network switch 103 preferably includes an Ethernet 10/100 connection.

The keypad 209 may be a contemporary keypad such as those used upon standard telephones. However, the keypad 209 may alternatively be any desired input device, such as a computer keyboard, a voice recognition system, a rotary dial, or any other desired input device.

The LCD display 210 is utilized to display the dialed number, as well as any other desired information such as network status, caller identification, etc.

Referring now to FIG. 3, the Internet Protocol switch controller 201 is shown in further detail. The Internet Protocol switch controller 201 comprises a bus 316 for facilitating

communication among va[illegible] portions thereof. The Internet Pro[illegible] switch controller 201 generally defines a network switch and has a switch engine queuing manager 301, a VLAN 802.1Q address look-up engine 302, an address resolution logic/virtual local area network (ARL/VLAN) table 303, an LED interface 304, a first 10/100 MAC 305, a second 10/100 MAC 306, a packet buffer 307, a 16-bit CPU interface 311 and a third 10/100 MAC 308, all in communication with the bus 316, so as to facilitate communication among one another.

The switch engine queuing manager 301 and the packet buffer 307 are configured to cooperate with one another so as to facilitate communication of network packets through the Internet Protocol switch controller 201 and to facilitate insertion of voice data packets onto the network without blocking of the network packets. An integrated address resolution unit is coupled to provide medium access control addresses and VLAN tag resolution at line rate. The integrated address resolution unit is configured to support ingress timestamp which in turn allows the queuing manager 301 to support egress delay flush. The integrated address resolution unit may configured to provide 256 medium access control addresses with a 16 bit tag.

The switch engine queuing manager 301 prioritizes packets going into the packet buffer 307. Thus, the switch engine queuing manager 301 defines a prioritization circuit. This prioritization applies to both voice data generated at the network telephone 101 by a user and voice data coming into the network telephone 101 from the network (if not already prioritized). Voice data may be recognized as such by the port it is received on (i.e., the MII port 203) by an identification header of the data packet which identifies the data contained within the packet as voce data or by associating either the source or destination of the packet as a voice device, e.g., a network telephone.

It should be appreciated that both non-voice data, such as that generated by the personal computer 102 and communicated to the network, as well as that provided by the network to the personal computer 102, passes through the network telephone 102 because of its serial connection between the personal computer 102 and the network switch 103. Thus, all such data must be processed by the Internet Protocol switch controller 201 of the network phone 101. It is important that voice data be given a higher priority than non-voice data, since such voice data is time critical, i.e., should pass through the network without unnecessary delay.

The avoidance of such delay is necessary so as to provide the desired degree of quality of service. As those skilled in the art will appreciate, a minimum threshold of quality of service is necessary in order to make use of the network telephone 101 worthwhile and desirable. The introduction of undesirable delays in voice packets transmitted over the network tends to cause a reduction in quality of service which may, if excessive, make use of the network telephone 101 undesirable.

The VLAN 802.1Q address look-up engine 302 utilizes the ARL/VLAN table 303 to facilitate desired routing of voice packets, as well as non-voice packets, across the network.

The LED interface 304 facilitates the use of LEDs 312 which may be utilized to provide any desired indication, such as transmit (TX), receive (RX) and status (network OK).

The search engine queuing manager 301 stores packets in the packet buffer 307 prior to the packets being sent to their destination, whether that destination is the voice engine processor 202 or the network. The packet buffer generally has a capacity greater than 200 Kbytes.

The 16-bit CPU interface 311 forms a portion of the microprocessor interface 204 which facilitates communication of control signals between the Internet Protocol switch controller 201 and the voice engine processor 202. The CPU interface 311 may be configured to facilitate the use of SNMP and BPDU frames. The CPU interface 311 may have counters coupled to provide RMON support.

The media independent interface 203 facilitates the communication of data, e.g., voice packets, between the Internet Protocol switch controller 201 and the voice engine processor 202. The media independent interface facilitates such communication via 10/100 MAC 308.

First media dependent interface 313 and second media dependent interface 314 facilitate communication with the personal computer 102 and the network switch 103, respectively. Media dependent interface 313 facilitates data communication via 10/100 Ethernet transceiver 309 and 10/100 MAC 305. Similarly, media dependent interface 314 facilitates data communication via 10/100 Ethernet transceiver 310 and 10/100 MAC 306.

The Internet Protocol switch VLAN tag frame format is generally defined in IEEE 802.3ac and 802.1Q. A four byte field is utilized and the tag is inserted between the source address (SA) and the original type-length field of the frame. The tag is split into two 16 bit fields. The first field is a VLAN tag protocol identified (TPID) and the second field is the tag control information (TCI), which contains the VLAN identifier (typically 12 bits), a 3-bit user priority field, and one CFI (Canonical Format Indicator) bit, which is generally not used in Ethernet networks.

Referring now to FIG. 4, the voice engine processor 202 comprises a CPU bus 420, which preferably operates at approximately 80 MHZ and a peripheral bus 421, which preferably operates at approximately 40 MHZ. An internal peripheral bridge 408 facilitates communication between the peripheral bus 421 and the CPU bus 420.

A CPU 401, preferably a RISC CPU having built-in cache such as a MIPS R3000, is in communication with the CPU bus 420 through DSP coprocessor 402. By using the DSP coprocessor 402, which cooperates with the CPU 401 the need for a off-chip DSP is eliminated.

At least one serial port 404, at least one general purpose input/output 405, a keyboard interface 406, at least one clock or timer 409, interrupt controller 410 and an LCD interface controller 407 are in communication with the peripheral bus 421. An on-chip PLL 440 may be used to provide a plurality of different clock speeds, e.g., 25 MHz, 33 MHz and 50 MHz.

A security module 411, a DMA arbiter 412, more than 20 Kbytes of SRAM buffer 413, a 14-bit CODE, a time division multiplexor interface 415 and a media independent interface 416 are in communication with the CPU bus 420. Also, a memory interface 403 which facilitates

interface to 100 MHZ SDR[illegible]417, SSRAM 418 and flash ROM 4[illegible] also in communication with the CPU bus 420.

The serial port 404 facilitates programming of the flash ROM 419, as well as debugging of the network telephone 101, particularly the operation of the voice engine processor 202 thereof.

The general purpose input/output(s) 405 allow the voice engine processor 202 to control external devices and/or to accept input from external devices, as desired.

The keyboard interface 406 facilitates electrical communication with keypad 209 and the LCD interface controller 407 facilitates communication with the LCD display 210.

The timer(s) 409 provide the clock signal(s) necessary for operation of the serial port(s) 404, GPIO(s) 405, keyboard interface 406, LCD interface controller 407 and interrupt controller 410. The interrupt controller 410 provides interrupts for the serial port(s) 404, GPIO(s) 405, keyboard interface 406 and LCD interface controller 407.

The GPIO(s) 405 facilitate debugging, the use of external devices such as LED indicators and other desired customer specific external logic.

The security module 411 applies encryption, preferably according to the defense encryption standard (DES) utilizing either single or triple DES and also facilitates decryption of encrypted data.

The DMA (Direct Memory Access) arbiter 412 controls access to the CPU bus 420 so as to facilitate direct memory access operations.

The SRAM buffer 413 provides temporary storage for voice data as it is processed by the security module 411 and/or the CPU 401 and the DSP co-processor 402.

In the transmit direction, CODEC 414 performs analog-to-digital conversion and filters the digital samples. In the receive direction, the CODEC 414 filters the received digital samples taken from the co-processor and then performs digital-to-analog conversion.

The DSP co-processor 402 is used to compress the voice data.

The CODEC 414 may be a G.711 CODEC and facilitates compression of voice data provided by the microphone 211 and facilitates decompression of voice data provided to the speaker 212. The CODEC 414 optionally also facilitates desired filtering of the voice packets provided by microphone 214, particularly so as to decrease the bandwidth necessary for transmission thereof.

Analog input circuit 430 receives an analog voice signal from the microphone 211 and conditions the signal for input to the CODEC 414. The analog input circuit 430 optionally filters and/or amplifies the analog voice signal before the analog voice signal is provided to the CODEC 414.

The analog output circuit 431 receives an analog signal from the CODEC 414 and provides the analog signal to speaker 212. The analog output circuit 431 optionally amplifies and/or filters the analog signal prior to providing the analog signal to the speaker 212.

The analog i[illegible]circuit 430 and the analog output circ[illegible]1 may be formed separate and external with respect to the voice engine processor 202, if the voice engine processor 202 is formed as a single chip.

The TDM interface 415 facilitates the handling of time division multiplexed data packets communicated upon the CPU bus 420 according to contemporary methodology.

The media independent interface 416 facilitates the communication of voice data packets between the voice engine processor 202 and the Internet Protocol switch controller 201.

The 100 MHZ SDRAM 417 provides a storage place for packetized voice data. The SSRAM 418 provides program memory for network related operations executed by the CPU 401. The flash ROM 419 is used to store program instructions for the RISC CPU 401 which are likely to need periodic updating.

As those skilled in the art will appreciate, the 14-bit DSP co-processor 402 may perform various different types of voice compression. For example, the CODEC 414 may utilize pulse code modulation (PCM), differential pulse code modulation (DPCM), adaptive differential pulse code modulation (ADPCM), motion picture experts group (MPEG)audio compression, linear predictive coding (LPC), code-excited linear prediction (CELP) and low-delay code-excited linear prediction (LD-CELP).

Of course, silence suppression is included in any type of audio compression. Those skilled in the art will appreciate that various other types of voice compression may likewise be suitable. The voice engine processor 202 is configured to provide echo control.

The voice engine processor 202 may optionally be configured to provide signaling for voice traffic, such as PBX voice traffic.

Referring now to FIG. 5, the processing of an incoming voice data packet by the network telephone 101 is shown. As shown in block 501, an incoming voice data packet is received by the network telephone 101. This incoming voice data packet is typically received via the media dependent interface 314 which facilitates communication with the network. Thus, the incoming voice data packet typically originates elsewhere upon the network, such as at another network telephone. However, such incoming voice data packets may alternatively originate at the personal computer 102 (such as by using the personal computer itself as a network telephone) or may originate at various network devices such as other computers (which are similarly utilized as network telephones) or servers which may have previously archived the voice data packet.

Regardless of the source of the incoming data packet. The address lookup engine will use the various DA and VLAN tag fields, as well as the ingress port identity (and possibly SA field), to determine which port(s) to queue the packet to, and whether to place it on the high or low priority queue(s) of the determined egress port(s). The priority tag of the voice data packet, if present, is read by the search engine queuing manager 301 of the IP switch controller 201 as shown in block 502. If the priority tag indicates that the packet contains voice data and therefore should be processed with high priority, the packet will be stored on the high priority queue.

Processing with high priority includes placing voice data packets ahead of other, lower priority packets in buffers, such that voice data packets are read from the buffers before other, non-voice data packets. In this manner, undesirable delays of the voice data packets are mitigated.

Thus, as shown in block 503, the voice data packet is stored by switch engine queuing manager 301 in packet buffer 307 for read-out from packet buffer 307 ahead of non-voice data packets and after any previously stored voice data packets. In this manner, voice data packets are read ahead of non-voice data packets and the order of voice data packets is maintained.

Since it is possible for voice data packets to arrive at the network telephone 101 out of order, the switch engine queuing manager 301 also places arriving packets in the packet buffer 307 in order, to the extent possible. The order of voice data packets is indicated in a header thereof. If a later transmitted voice data packet arrives before an earlier transmitted voice data packet, then the earlier transmitted voice data packet is placed in the voice data packet buffer 307 ahead of the later transmitted voice data packet. In this manner, the switch engine queuing manager 301 attempts to maintain desired order of the voice data packets, such that undesirable disruptions of received speech are mitigated. Of course, if an earlier transmitted voice data packet arrives after a later transmitted voice data packet has already been read from packet buffer 307 (and has generally already been listened to via speaker 212), then the earlier transmitted voice data packet must be deleted, e.g., not stored in packet buffer 307, so as to mitigate further degradation of the incoming voice data stream.

If the address lookup 302 determines the packet is destined for internet telephone 101, it will queue the packet to the MII port 203, and directs the queuing manager 301 to place it on the high priority queue for that port, assuming its priority field indicated this. After the voice data packets have been queued ahead of non-voice data packets and stored in order, to the extent possible, the voice data packets are communicated from the packet buffer 307 to SRAM buffer 413 of the voice engine processor 202 via media independent interface 203, as show in block 504. Voice data packets may also be stored in the 100 MHZ SDRAM, as necessary to prevent over flowing of the SRAM buffer 413. The voice engine processor processes the voice data packets so as to provide a voice signal compatible with speaker 212 to facilitate listening to the received voice signal. Such processing is performed by the CPU 401 and the DSP coprocessor 402 by executing instructions stored in the SSRAM 418 and flash ROM 419 while operating upon voice data packets stored in SRAM buffer 413. Additionally, voice data packets are transferred from the 100 MHZ SDRAM 417 to the SRAM buffer 413 as necessary to facilitate further processing thereof.

As shown in block 505, the CPU 401 of the voice engine processor 202 facilitates de-packetizing of the voice data packets stored in SRAM buffer 413.

As shown in block 506, if the voice data packets are encrypted, then the security module 401 decrypts the voice data packets.

As shown in block 507, DSP co-processor 402 decompresses packetized voice data. If pulse code modulation (PCM) only is used, then such decompression inherently results in digital-to-analog conversion of the voice signal. However, if more sophisticated voice compression is utilized, then CODEC 414 performs decompression of the voice signal and then performs digital-to-analog conversion thereof, as shown in block 508.

As shown in block 509, the decompressed analog signal is provided to analog output 431 which conditions, e.g., filters, amplifies and/or attenuates, the signal such that the signal is suitable for speaker 212.

Referring now to FIG. 6, the processing of outgoing voice data from the network telephone 101 is generally the reverse of the processing of incoming voice data, as shown in FIG. 5.

More particularly, as shown in block 601 an analog voice signal which results from talking into microphone 211 is communicated from microphone 211 to voice engine processor 202 via analog input circuit 430. The analog input circuit 430 conditions the output of microphone 211 so as to be suitable for processing by CODEC 414 of the voice engine processor 202. Thus, the analog input circuit 430 filters, amplifies and/or attenuates the analog voice signal communicated from the microphone.

As shown in block 602, CODEC 414 performs analog-to-digital conversion on the analog voice signal and stores the digital voice signal in SRAM buffer 413. If only PCM voice compression is utilized, then the analog-to-digital conversion process results in a compressed digital voice signal.

As shown in block 603, if more sophisticated compression is utilized, then DSP co-processor 402 compresses the digitized voice signal, utilizing such voice compression.

As shown in block 604, security module 411 encrypts the compressed voice signal, if desired. Defense encryption standard (DES) or triple DES are utilized. However, those skilled in the art will appreciate that various other types of encryption are likewise suitable and may either alternatively or additionally be utilized.

As shown in block 605, CPU 401 packetizes the voice signal, such that the voice signal is suitable for transmission over a network, such as the Internet.

As shown in block 606, the voice data packet is communicated from the SRAM buffer 413 and/or 100 MHZ SDRAM to the packet buffer 307 of the IP switch controller 201, via media independent interface 203.

The address lookup engine 302 uses the DA, VLAN tag and ingress port to determine the egress port(s) to which the frame must be queued, and informs the switch engine queuing manager 301 of this decision, as well as whether to queue as high or low priority. In general, all packets originating from internet telephone 101 will be tagged as high priority.

As shown in block 607, a priority tag is applied to the voice data packet by the search engine queuing manager 301 of the IP switch controller 201. Routing information may also be

applied to the voice data pack[illegible]the queuing manager, in accordance [illegible] the instructions from the VLAN 802.1Q address lookup engine 302 and the ARL/VLAN table 303.

As shown in block 608, the voice data packet is stored by switch engine queuing manager 301 and packet buffer 307 for read-out from packet buffer 307 ahead of non-voice data packets (which may have arrived from either the network or personal computer 102) and after any previously stored voice data packets. In this manner, the priority necessary to maintain the desired quality of service and to prevent undesirable degradation of the voice signal is provided.

It is worthwhile to note that other data packets, besides those originating at the network telephone 101 may be stored in packet buffer 307 and then transmitted to the network. For example, data packets provided by personal computer 102 must be transmitted, without detrimental alteration, to switch 103 of the network. The switch engine queuing manager 301 may, optionally, apply a desired priority tag to such non-voice data packets which originated at the personal computer 102. These non-voice data packets will be assigned a priority lower than that of the voice data packets which originate at the network telephone 101.

As shown in block 609, voice data packets are transmitted from the network telephone to the network, via media dependent interface 314, which typically facilitates transmission of the voice data packets to switch network 103 or any other desired network device. The voice data packets are then routed by switch 103 using routing information, such as that applied by the VLAN address look-up table 302 to the voice data packets according to contemporary methodology.

Referring now to FIG. 7, the application of priority tags to data packets passing through the network telephone 101 is shown. The priority tags are applied by the switch engine queuing manager 301 prior to storing the data packets in packet buffer 307.

As shown in block 701, packets are received by the IP switch controller 201 of the network telephone 101. These packets may be received from the network, the personal computer 102 or the voice engine processor 202 (which occurs when the packets originate at the network telephone 101 and are the result of speech from the microphone 211, or entries on the keypad 209, being processed by voice engine processor 202).

As shown in block 702, high priority packets are processed differently from low priority packets. High priority packets are voice data packets and low priority packets are non-voice data packets. If the packet received by the IP switch controller 201 is a non-voice data packet (as indicated by the packet header or routing information), then processing is transferred to block 703, wherein the switch engine queuing manager 301 stores the packet for low priority use in the packet buffer 307. Storing the packet for low priority use may be accomplished by storing the packet in a dedicated low priority portion of the packet buffer 307, may comprise maintaining a table of the storage location of the low priority packet such that the packet may later be recognized as a low priority packet or may include the application of a priority tag to the packet, such that the packet is readily recognized as a low priority packet.

As shown in block 704, when the packet is recognized as a high priority packet, i.e., a voice data packet, by block 702, then the switch engine queuing manager 301 stores the packet for high priority use in packet buffer 307. Similarly, the high priority packet may be stored in a dedicated high priority portion of the packet buffer 307, a table may be maintained designating where high priority packets are stored within the packet buffer 307, or a tag may be applied to the packet such that the packet may be readily identified as a high priority packet.

As shown in block 705, packets are read from the packet buffer 307 in order, according to priority of the packets. That is, high priority packets are read from the packet buffer 307, generally in order of their reception, and then low priority packets are read from the packet buffer 307. Thus, higher priority voice data packets are read from the packet buffer 307 and are processed prior to reading lower priority non-voice data packets from the packet buffer 307 and processing the lower priority non-voice data packets.

Note that while FIG. 7 shows only two levels of priority, high and low, the 3-bit user priority field in the 802.3ac/802.1Q tag allows eight levels of priority to be encoded, and the IP switch could be extended to support these multiple priority queues.

It is understood that the exemplary network telephone system described herein and shown in the drawings represents only a presently preferred embodiment of the invention. Indeed, various modifications and additions may be made to such embodiment without departing from the spirit and scope of the invention.

CLAIMS

1. A network telephone comprising:

a microphone coupled to provide voice data to a network;

a speaker configured to facilitate listening to voice data from the network;

a dialing device coupled to facilitate routing of voice data upon the network;

a first port configured to facilitate communication with a first network device;

a second port configured to facilitate communication with a second network device; and

a prioritization circuit coupled to apply prioritization to voice data provided by the microphone.

2. The network telephone as recited in claim 1, wherein the first port is configured to facilitate communication of voice data packets with the first network device and the second port is configured to facilitate communication of voice data packets with the second network device.

3. The network telephone as recited in claim 1, wherein the microphone and the speaker at least partially define a handset.

4. The network telephone as recited in claim 1, wherein the dialing device comprises a keypad.

5. The network telephone as recited in claim 1, wherein the first port and the second port comprise Ethernet 10/100 ports.

6. The network telephone as recited in claim 1, wherein the prioritization circuit is defined by a network switch.

7. The network telephone as recited in claim 1, wherein the prioritization circuit is defined by an Ethernet switch.

8. The network telephone as recited in claim 1, wherein:

the prioritization circuit is defined by a network switch; and

further comprising a voice engine processor in communication with the network switch, the voice engine processor being configured to digitize and compress voice data from the microphone and to decompress and perform digital to analog conversion upon voice data provided to the speaker.

9. The network telephone as recited in claim 1, wherein:

the prioritization circuit is defined by a network switch; and

further comprising a voice engine processor in communication with the network switch, the voice engine processor being configured to digitize, compress and packetize voice data from the microphone and to depacketize, decompress and perform digital to analog conversion upon voice data provided to the speaker.

10. The network telephone as recited in claim 1, wherein the prioritization circuit is configured to tag voice data packets to facilitate prioritization thereof.

11. The network telephone as recited in claim 1, wherein the prioritization circuit is configured to tag voice data packets to facilitate prioritization thereof and is configured to read tags on data packets provided thereto by the network to facilitate prioritization thereof.

12. A network telephone comprising:

a switch controller having at least one port for facilitating electrical communication with a network; and

a voice engine processor in electrical communication with the switch controller, the voice processor having a microphone port for facilitating electrical communication with a microphone and having a speaker port for facilitating electrical communication with a speaker.

13. The telephone as recited in claim 12, wherein the switch controller is configured to apply prioritization to voice packets.

14. The telephone as recited in claim 12, wherein the switch controller is configured to apply prioritization to voice data packets and to route voice data packets over a network.

15. The telephone as recited in claim 12, wherein the switch controller is configured to apply prioritization to voice data packets and to route voice data packets over an Ethernet.

16. The telephone as recited in claim 12, wherein the switch controller is configured to apply prioritization to voice data packets and to route voice data packets over the Internet.

17. The telephone as recited in claim 12, wherein the switch controller is configured to apply prioritization to voice data packets provided by the microphone and coupled to route the voice data packets over a network.

18. The network [illegible]hone as recited in claim 12, where[illegible]e switch controller is configured to tag voice data packets to facilitate prioritization thereof.

19. The network telephone as recited in claim 12, wherein the switch controller is configured to tag voice data packets to facilitate prioritization thereof and is configured to read tags on data packets provided thereto by the network to facilitate prioritization thereof.

20. The telephone as recited in claim 12, wherein the switch controller is configured to be compatible with Internet Protocol.

21. The telephone as recited in claim 12, wherein electrical communication between the switch controller and the voice engine processor is facilitated via a media independent interface and a microprocessor interface.

22. The telephone as recited in claim 12, wherein the switch controller comprises two ports for facilitating communication with the network.

23. The telephone as recited in claim 12, wherein the switch controller comprises two Ethernet ports for facilitating communication with the network.

24. The telephone as recited in claim 12, wherein the switch controller comprises two 10/100 megabit/sec Ethernet ports for facilitating communication with the network.

25. The telephone as recited in claim 12, wherein the voice engine processor further comprises a keypad port for facilitating communication with a keypad.

26. The telephone as recited in claim 12, wherein the voice engine processor further comprises a display port for facilitating communication with a display.

27. The telephone as recited in claim 12, wherein the switch controller is configure to be placed serially into a Ethernet transmission medium intermediate a network interface card and a switch.

28. The telephone as recited in claim 12, wherein the voice engine processor is configured to compress voice communications.

29. The telephone as recited in claim 12, wherein the voice engine processor is configured to compress voice communications using PCM compression.

30. The telephone as recited in claim 12, wherein the voice engine processor is configured to suppress silence.

31. The telephone as recited in claim 12, wherein the voice engine processor is configured to provide a desired level of quality of service.

32. The telephone as recited in claim 12, wherein the voice engine processor is configured to provide signaling for voice traffic.

33. The telephone as recited in claim 12, wherein the voice engine processor is configured to provide signaling for PBX voice traffic.

34. The telephone as recited in claim 12, wherein the voice engine processor is configured to provide echo control.

35. A network telephone comprising a prioritization circuit coupled to tag voice data packets with information representative of a priority thereof and coupled to read tags associated with packets.

36. A switch controller for controlling a network switch through which voice data is communicated, the switch controller comprising:

a packet buffer coupled to buffer voice data packets and non-voice data packets;

a switch engine queuing manager coupled to queue voice data packets and non-voice data packets in the packet buffer in a manner which enhances quality of service for the voice data packets;

at least one transceiver in communication with the packet buffer;

a network interface coupled facilitate communication between each transceiver and a network; and

a voice interface coupled to facilitate communication between a voice source and the packet buffer.

37. The switch controller as recited in claim 36, wherein the transceiver comprises a 10/100 transceiver.

38. The switch controller as recited in claim 36, further comprising a medium access control for each transceiver coupled to facilitate communication of voice packets and non-voice packets between the packet buffer and a network.

39. The switch controller as recited in claim 36, further comprising a 10/100 medium access control for each transceiver.

40. The switch controller as recited in claim 36, further comprising a medium access control coupled to facilitate communication of voice data packets between the voice source and the packet buffer via the voice interface.

41. The switch controller as recited in claim 36, further comprising a media dependent interface coupled to facilitate communication between each transceiver and the network.

42. The switch controller as recited in claim 36, further comprising a media independent interface coupled to facilitate communication between the packet buffer and the voice source.

43. The switch controller as recited in claim 36, further comprising a CPU interface coupled to facilitate communication between a CPU and the search engine queuing manager.

44. The switch controller as recited in claim 36, further comprising a CPU interface coupled to facilitate communication between a CPU and the search engine queuing manager, the CPU interface facilitating use of SNMP and BPDU frames.

45. The switch controller as recited in claim 36, further comprising a CPU interface coupled to facilitate communication between a CPU and the search engine queuing manager, the CPU interface having counters coupled to provide RMON support.

46. The switch controller as recited in claim 36, further comprising:

an address table; and

an address lookup engine couple to fetch addresses from the address table and to provide the addresses to the transceiver(s) to facilitate network routing.

47. The switch controller as recited in claim 36, further comprising:

a VLAN address table; and

an VLAN address lookup engine couple to fetch addresses from the address table and to provide the addresses to the transceiver(s) to facilitate network routing.

48. The switch controller as recited in claim 36, further comprising:

a display coupled to indicated a status of the switch controller; and

a display interface coupled to communicate status information to the display.

49. The s[illegible] controller as recited in claim 36, fu[illegible] comprising:
a plurality of LEDs coupled to indicated a status of the switch controller; and
an LED interface coupled to communicate status information to the LEDs.

50. The switch controller as recited in claim 36, wherein the switch engine queuing manager and the packet buffer are configured to cooperate so as to facilitate communication of network packets through the switch controller and to facilitate insertion of voice packets onto the network without blocking of the network packets.

51. The switch controller as recited in claim 36, further comprising an integrated address resolution unit coupled to provide medium access control addresses and VLAN tag resolution.

52. The switch controller as recited in claim 36, further comprising an integrated address resolution unit coupled to provide medium access control addresses and VLAN tag resolution at line rate.

53. The switch controller as recited in claim 36, further comprising an integrated address resolution unit configured to support ingress timestamp and egress delay flush.

54. The switch controller as recited in claim 36, further comprising an integrated address resolution unit configured to provide 256 medium access control addresses with a 16 bit tag.

55. A voice engine processor for use in network telephony, the voice engine processor comprising:
a CODEC;
a input port coupled to communicate an input audio signal from an input transducer to the CODEC;
an output port coupled to communicate an output audio signal from the CODEC to an output transducer;
a CPU; and
a DSP coprocessor coupled to the CPU so as to facilitate digital signal processing of voice data.

56. The voice engine as recited in claim 55, further comprising a microprocessor interface coupled to facilitate communication between the CPU and a switch controller.

57. The voice engine as recited in claim 55, wherein the CODEC comprises a 14 bit CODEC.

58. The voice engine as recited in claim 55, wherein the input port is configured to communicate a microphone output signal to the CODEC.

59. The voice engine as recited in claim 55, wherein the output port is configured to communicate an audio signal to a speaker.

60. The voice engine as recited in claim 55, further comprising:
a CPU;
memory in communication with the CPU;
a buffer; and
wherein the CPU is responsive to instructions stored in the memory so as to effect storage of information representative of at least one of the input audio signal and the output audio signal in the buffer.

61. The voice engine as recited in claim 55, further comprising:
a CPU;
memory in communication with the CPU;
a buffer in communication with the CPU; and
a digital signal coprocessor in communication with the CPU.

62. The voice engine as recited in claim 55, further comprising:
a serial port in communication with the CPU;
a general purpose input/output port in communication with the CPU;
a keyboard port in communication with the CPU; and
an LCD controller in communication with the CPU.

63. The voice engine as recited in claim 55, further comprising:
a serial port in communication with the CPU;
a general purpose input/output port in communication with the CPU;
a keyboard port in communication with the CPU;
an LCD controller in communication with the CPU;
a clock in communication with the CPU; and
an interrupt controller in communication with the CPU.

64. The v[illegible]engine as recited in claim 55, furthe[illegible]nprising:
a peripheral bus in communication with the CPU via an IPB bridge;
a serial port in communication with the peripheral bus;
a general purpose input/output port in communication with the peripheral bus;
a keyboard port in communication with the peripheral bus;
an LCD controller in communication with the peripheral bus;
a clock in communication with the peripheral bus; and
an interrupt controller in communication with the peripheral bus.

65. The voice engine as recited in claim 55, further comprising:
a security module in communication with the CPU;
a DMA buffer in communication with the CPU;
a TDM interface in communication with the CPU; and
a MII interface in communication with the CPU.

66. The voice engine as recited in claim 55, further comprising:
a CPU bus in communication with the CPU;
a security module in communication with the CPU bus;
a DMA buffer in communication with the CPU bus;
a TDM interface in communication with the CPU bus; and
a MII interface in communication with the CPU bus.

67. The voice engine as recited in claim 55, further comprising:
a memory interface in communication with the CPU;
SDRAM in communication with the memory interface;
SSRAM in communication with the memory interface; and
flash ROM in communication with the memory interface.

68. A network telephone system comprising:
a network; and
at least one network telephone, each network telephone comprising a network switch coupled to apply prioritization to voice packets and coupled to read prioritization of voice packets.

69. A method for communicating voice via a network, the method comprising:
facilitating routing of voice data upon the network via a dialing device;
providing voice data to a network via a microphone;
listening to voice data from the network via a speaker;

facilitating communication with a first network device via a first port;
facilitating communication with a second network device via a second port; and
prioritizing voice data provided by the microphone.

70. The method as recited in claim 69, wherein the first port is configured to facilitate communication of voice packets with the first network device and the second port is configured to facilitate communication of voice packets with the second network device.

71. The method as recited in claim 69, wherein the microphone and the speaker comprise a handset.

72. The method as recited in claim 69, wherein the dialing device comprises a keypad.

73. The method as recited in claim 69, wherein the first port and the second port comprise Ethernet 10/100 ports.

74. The method as recited in claim 69, wherein the prioritization circuit is defined by a network switch.

75. The method as recited in claim 69, wherein the prioritization circuit is defined by an Ethernet switch.

76. The method as recited in claim 69, wherein:
the prioritization circuit is defined by a network switch; and
further comprising a voice engine processor in communication with the network switch, the voice engine processor being configured to digitize and compress voice data from the microphone and to decompress and perform digital to analog conversion upon voice data provided to the speaker.

77. The method as recited in claim 69, wherein:
the prioritization circuit is defined by a network switch; and
further comprising a voice engine processor in communication with the network switch, the voice engine processor being configured to digitize, compress voice and packetize data from the microphone and to depacketize, decompress and perform digital to analog conversion upon voice data provided to the speaker.

78. The method as recited in claim 69, wherein the prioritization circuit is configured to tag voice packet to facilitate prioritization thereof.

79. The method as recited in claim 69, wherein the prioritization circuit is configured to tag voice packet to facilitate prioritization thereof and is configured to read tags on data packets provided thereto by the network to facilitate prioritization thereof.

80. A method comprising:

using a switch controller having at least one port to facilitate electrical communication with a network; and

using a voice engine processor in electrical communication with the switch controller to process voice, the voice processor having a microphone port for facilitating electrical communication with a microphone and having a speaker port for facilitating electrical communication with a speaker.

81. The method as recited in claim 80, wherein the switch controller is configured to apply prioritization to voice packets.

82. The method as recited in claim 80, wherein the switch controller is configured to apply prioritization to voice packets and to route voice packets over a network.

83. The method as recited in claim 80, wherein the switch controller is configured to apply prioritization to voice packets and to route voice packets over an Ethernet.

84. The method as recited in claim 80, wherein the switch controller is configured to apply prioritization to voice packets and to route voice packets over the Internet.

85. The method as recited in claim 80, wherein the switch controller is configured to apply prioritization to voice packets provided by the microphone and coupled to route the vice packets over a network.

86. The method as recited in claim 80, wherein the switch controller is configured to tag voice packet to facilitate prioritization thereof.

87. The method as recited in claim 80, wherein the switch controller is configured to tag voice packet to facilitate prioritization thereof and is configured to read tags on data packets provided thereto by the network to facilitate prioritization thereof.

88. The method as recited in claim 80, wherein the switch controller is configured to be compatible with Internet Protocol.

89. The method as recited in claim 80, wherein electrical communication between the switch controller and the voice engine processor is facilitated via a media independent interface and a microprocessor interface.

90. The method as recited in claim 80, wherein the switch controller comprises two ports for facilitating communication with the network.

91. The method as recited in claim 80, wherein the switch controller comprises two Ethernet ports for facilitating communication with the network.

92. The method as recited in claim 80, wherein the switch controller comprises two 10/100 megabit/sec Ethernet ports for facilitating communication with the network.

93. The method as recited in claim 80, wherein the voice engine processor further comprises a keypad port for facilitating communication with a keypad.

94. The method as recited in claim 80, wherein the voice engine processor further comprises a display port for facilitating communication with a display.

95. The method as recited in claim 80, wherein the switch controller is configure to be place serially into a Ethernet transmission medium intermediate a network interface card and a switch.

96. The method as recited in claim 80, wherein the voice engine processor is configured to compress voice communications.

97. The method as recited in claim 80, wherein the voice engine processor is configured to compress voice communications using PCM compression.

98. The method as recited in claim 80, wherein the voice engine processor is configured to suppress silence.

99. The method as recited in claim 80, wherein the voice engine processor is configured to provide a desired level of quality of service.

100. The method as recited in claim 80, wherein the voice engine processor is configured to provide signaling for voice traffic.

101. The method as recited in claim 80, wherein the voice engine processor is configured to provide signaling for PBX voice traffic.

102. The method as recited in claim 80, wherein the voice engine processor is configured to provide echo control.

103. A method for communicating voice via a network, the method comprising tagging voice packets with information representative of a priority thereof and reading tags associated with packets.

104. A method for controlling a network switch through which voice data is communicated, the method comprising:

buffering voice packets and non-voice packets;

queuing voice packets and non-voice packets in the packet buffer in a manner which enhances quality of service for the voice packets;

communicating voice packets from the packet buffer to at least one transceiver;

communicating voice packets between each transceiver and a network; and

communicating between a voice source and the packet buffer.

105. The method as recited in claim 104, wherein the transceiver comprises a 10/100 transceiver.

106. The method as recited in claim 104, further comprising a medium access control for each transceiver coupled to facilitate communication of voice packets and non-voice packets between the packet buffer and a network.

107. The method as recited in claim 104, further comprising a 10/100 medium access control for each transceiver.

108. The method as recited in claim 104, further comprising a medium access control coupled to facilitate communication of voice packets between the voice source and the packet buffer via the voice interface.

109. The method as recited in claim 104, further comprising a media dependent interface coupled to facilitate communication between each transceiver and the network.

110. The method as recited in claim 104, further comprising a media independent interface coupled to facilitate communication between the packet buffer and the voice source.

111. The method as recited in claim 104, further comprising a CPU interface coupled to facilitate communication between a CPU and the search engine queuing manager.

112. The method as recited in claim 104, further comprising a CPU interface coupled to facilitate communication between a CPU and the search engine queuing manager, the CPU interface facilitating use of SNMP and BPDU frames.

113. The method as recited in claim 104, further comprising a CPU interface coupled to facilitate communication between a CPU and the search engine queuing manager, the CPU interface having counters coupled to provide RMON support.

114. The method as recited in claim 104, further comprising:
an address table; and
an address lookup engine couple to fetch addresses from the address table and to provide the addresses to the transceiver(s) to facilitate network routing.

115. The method as recited in claim 104, further comprising:
a VLAN address table; and
an VLAN address lookup engine couple to fetch addresses from the address table and to provide the addresses to the transceiver(s) to facilitate network routing.

116. The method as recited in claim 104, further comprising:
a display coupled to indicated a status of the switch controller; and
a display interface coupled to facilitate communicate status information to the display.

117. The method as recited in claim 104, further comprising:
a plurality of LEDs coupled to indicated a status of the switch controller; and
an LED interface coupled to facilitate communicate status information to the LEDs.

118. The method as recited in claim 104, wherein the switch engine queuing manager and the packet buffer are configured to cooperate so as to facilitate communication of network packets through the switch controller and to facilitate insertion of voice packets onto the network without blocking of the network packets.

119. The method as recited in claim 104, further comprising an integrated address resolution unit coupled to provide medium access control addresses and VLAN tag resolution.

120. The method as recited in claim 104, further comprising an integrated address resolution unit coupled to provide medium access control addresses and VLAN tag resolution at line rate.

121. The method as recited in claim 104, further comprising an integrated address resolution unit configured to support ingress timestamp and egress delay flush.

122. The method as recited in claim 104, further comprising an integrated address resolution unit configured to provide 256 medium access control addresses with a 16 bit tag.

123. A method for processing voice for communication via a network, the method comprising:

communicating an analog input audio signal from an input transducer to an analog to digital converter to provide a digital input audio signal;

communicating a digital output audio signal to a digital to analog converter to provide an analog output audio signal to an output transducer; and

processing the digital input audio signal and the digital output audio signal with a DSP coprocessor coupled to the CPU.

124. The method as recited in claim 123, further comprising a microprocessor interface coupled to facilitate communication between the CPU and a switch controller.

125. The method as recited in claim 123, wherein the CODEC comprises a 14 bit CODEC.

126. The method as recited in claim 123, wherein the input port is configured to communicate a microphone output signal to the CODEC.

127. The method as recited in claim 123, wherein the output port is configured to communicate an audio signal to a speaker.

128. The method as recited in claim 123, further comprising:

a CPU;

memory in communication with the CPU;

a buffer; and

wherein the CPU is responsive to instructions stored in the memory so as to effect storage of information representative of at least one of the input audio signal and the output audio signal in the buffer.

129. The method as recited in claim 123, further comprising:
a CPU;
memory in communication with the CPU;
a buffer in communication with the CPU; and
a digital signal coprocessor in communication with the CPU.

130. The method as recited in claim 123, further comprising:
a serial port in communication with the CPU;
a general purpose input/output port in communication with the CPU;
a keyboard port in communication with the CPU; and
an LCD controller in communication with the CPU.

131. The method as recited in claim 123, further comprising:
a serial port in communication with the CPU;
a general purpose input/output port in communication with the CPU;
a keyboard port in communication with the CPU;
an LCD controller in communication with the CPU;
a clock in communication with the CPU; and
an interrupt controller in communication with the CPU.

132. The method as recited in claim 123, further comprising:
a peripheral bus in communication with the CPU via an IPB bridge:
a serial port in communication with the peripheral bus;
a general purpose input/output port in communication with the peripheral bus;
a keyboard port in communication with the peripheral bus;
an LCD controller in communication with the peripheral bus;
a clock in communication with the peripheral bus; and
an interrupt controller in communication with the peripheral bus.

133. The method as recited in claim 123, further comprising:
a security module in communication with the CPU;
a DMA buffer in communication with the CPU;
a TDM interface in communication with the CPU; and
a MII interface in communication with the CPU.

134. The method as recited in claim 123, further comprising:
a CPU bus in communication with the CPU;
a security module in communication with the CPU bus;

a DMA buff[illegible] communication with the CPU bus;
a TDM interface in communication with the CPU bus; and
a MII interface in communication with the CPU bus.

135. The method as recited in claim 123, further comprising:
a memory interface in communication with the CPU;
SDRAM in communication with the memory interface;
SSRAM in communication with the memory interface; and
flash ROM in communication with the memory interface.

136. A method for communicating voice comprising:
providing a network; and
applying prioritization to voice packets and reading prioritization of voice packets via at least one network telephone, each network telephone comprising a network switch.

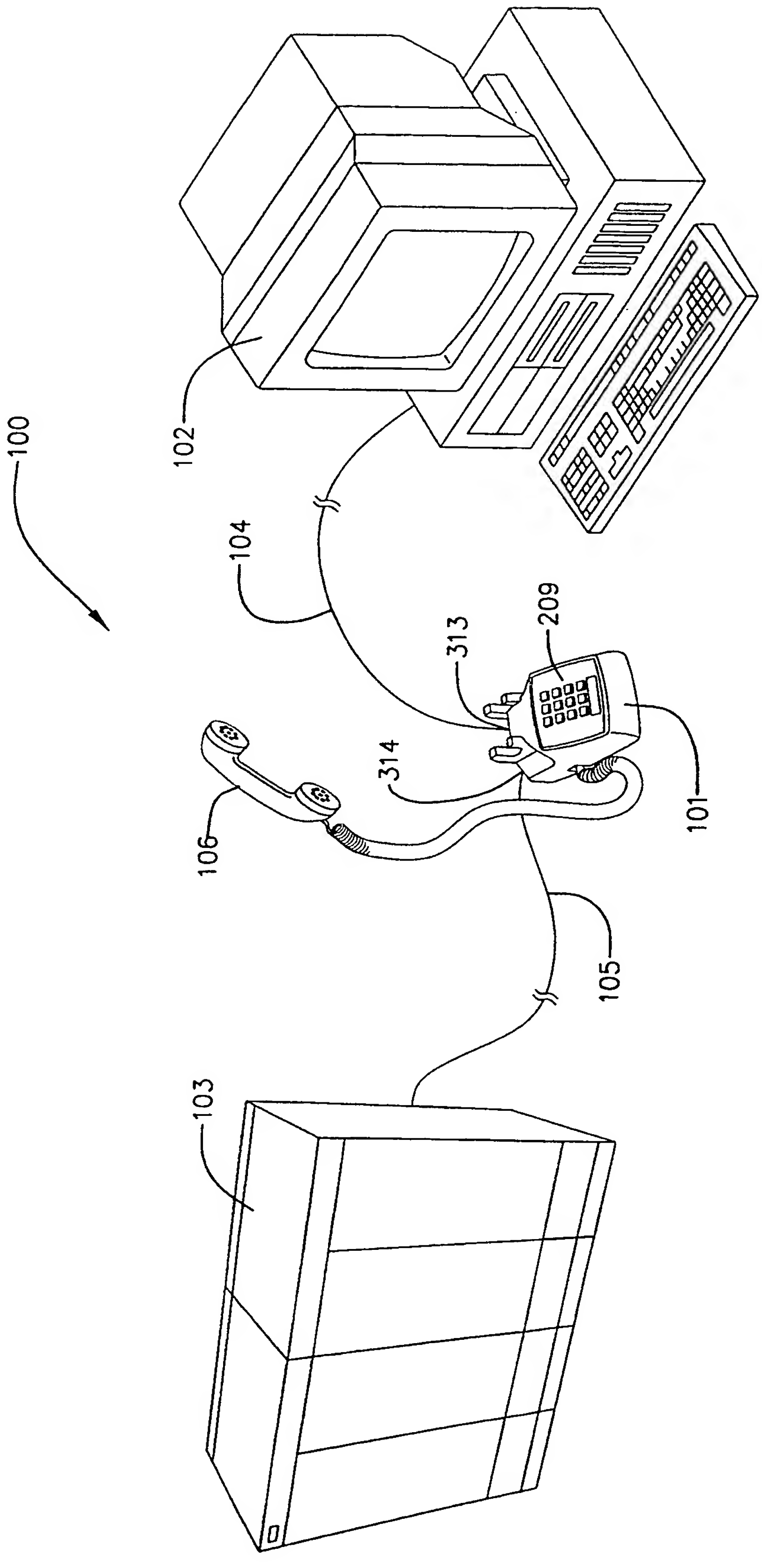
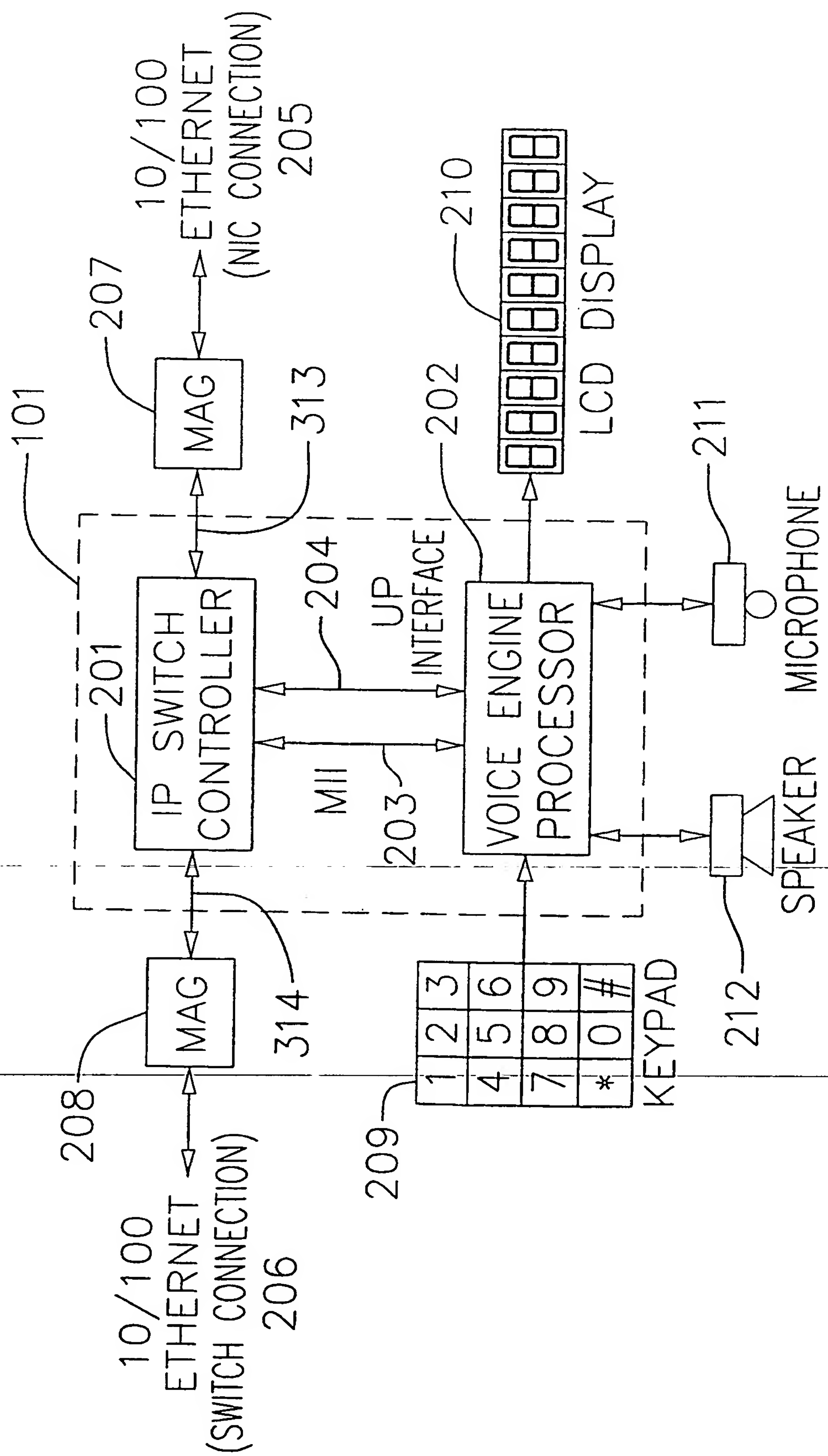


FIG.1

FIG. 2


10/100
ETHERNET
(NIC CONNECTION)
205
207
MAG
313
101
IP SWITCH
CONTROLLER
201
204
UP
INTERFACE
MII
203
202
VOICE ENGINE
PROCESSOR
210
LCD DISPLAY
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SPEAKER
212
314
MAG
208
10/100
ETHERNET
(SWITCH CONNECTION)
206
KEYPAD
209

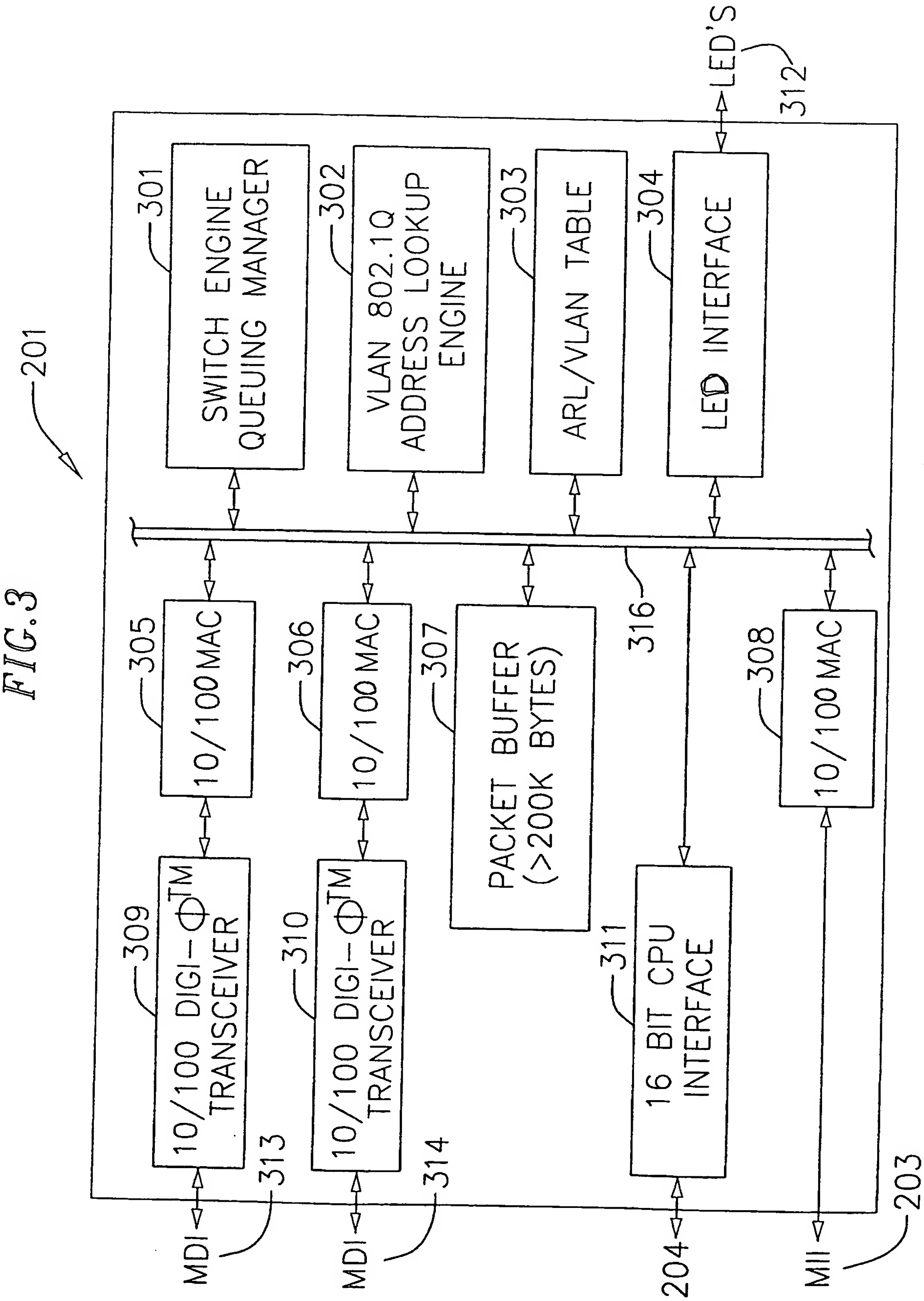

201
SWITCH ENGINE QUEUING MANAGER
301
VLAN 802.1Q ADDRESS LOOKUP ENGINE
302
ARL/VLAN TABLE
303
LED INTERFACE
304
LED'S
312
10/100 MAC
305
10/100 MAC
306
PACKET BUFFER (>200K BYTES)
307
316
10/100 MAC
308
10/100 DIGI-Φ™ TRANSCEIVER
309
10/100 DIGI-Φ™ TRANSCEIVER
310
16 BIT CPU INTERFACE
311
MDI
313
MDI
314
204
MII
203

FIG. 3

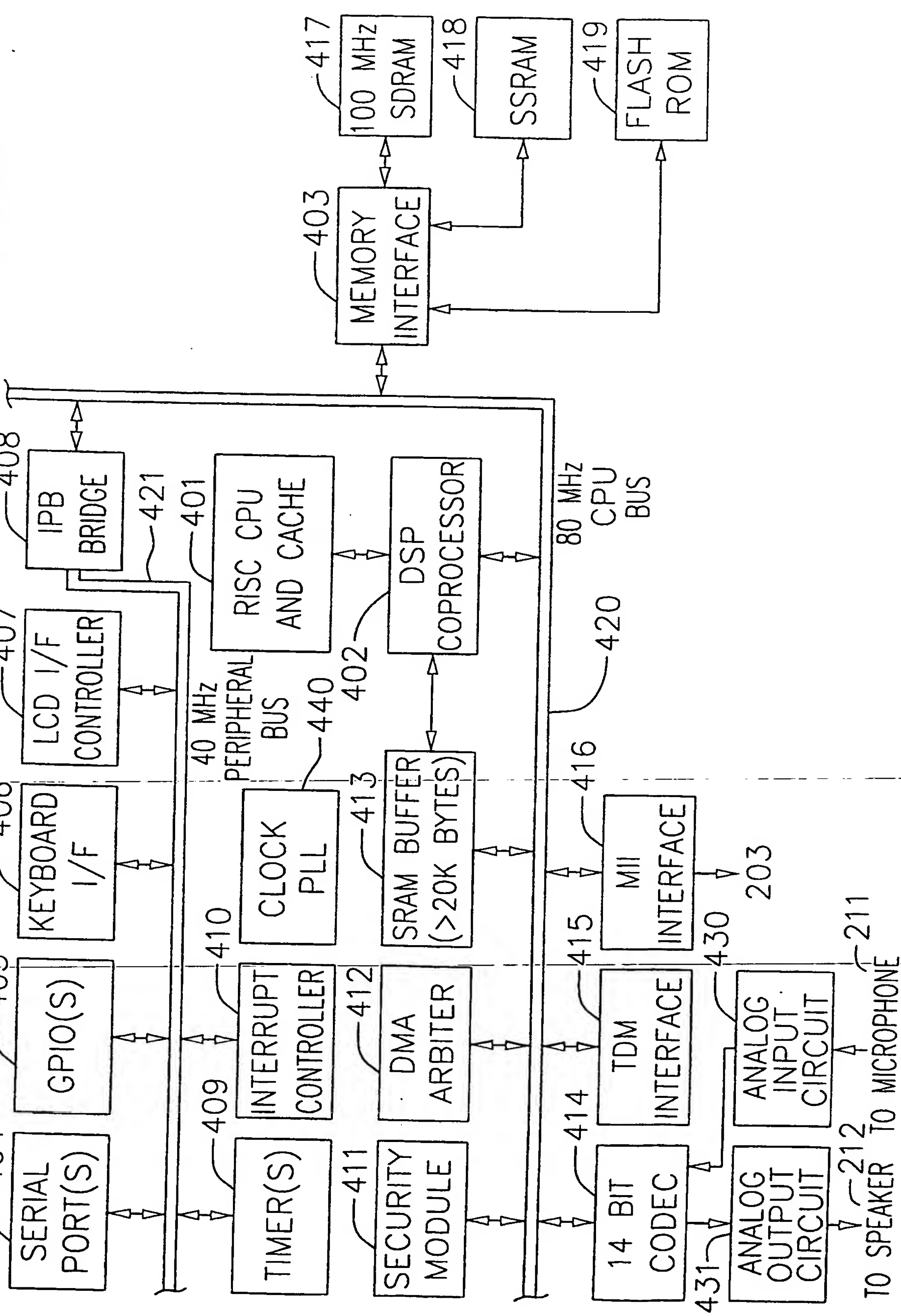

202
408
IPB BRIDGE
421
401
RISC CPU AND CACHE
DSP COPROCESSOR
80 MHz CPU BUS
403
MEMORY INTERFACE
417
100 MHz SDRAM
418
SSRAM
419
FLASH ROM
407
LCD I/F CONTROLLER
40 MHz PERIPHERAL BUS
440
402
420
406
KEYBOARD I/F
CLOCK PLL
413
SRAM BUFFER (>20K BYTES)
416
MII INTERFACE
203
405
GPIO(S)
410
INTERRUPT CONTROLLER
412
DMA ARBITER
415
TDM INTERFACE
430
ANALOG INPUT CIRCUIT
211
TO MICROPHONE
404
SERIAL PORT(S)
409
TIMER(S)
411
SECURITY MODULE
414
14 BIT CODEC
431
ANALOG OUTPUT CIRCUIT
212
TO SPEAKER

FIG. 4

FIG.5

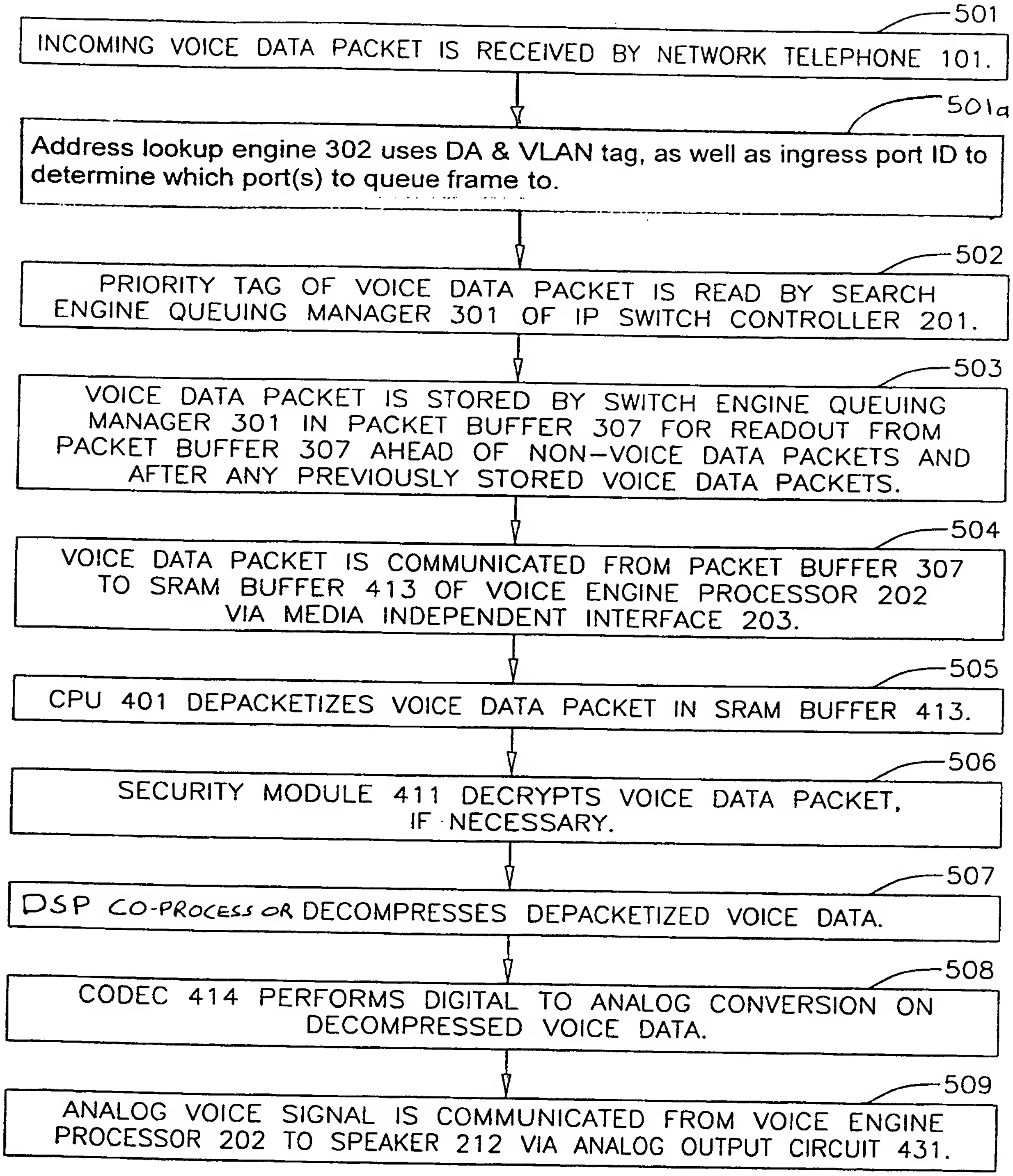

501
INCOMING VOICE DATA PACKET IS RECEIVED BY NETWORK TELEPHONE 101.
501a
Address lookup engine 302 uses DA & VLAN tag, as well as ingress port ID to determine which port(s) to queue frame to.
502
PRIORITY TAG OF VOICE DATA PACKET IS READ BY SEARCH ENGINE QUEUING MANAGER 301 OF IP SWITCH CONTROLLER 201.
503
VOICE DATA PACKET IS STORED BY SWITCH ENGINE QUEUING MANAGER 301 IN PACKET BUFFER 307 FOR READOUT FROM PACKET BUFFER 307 AHEAD OF NON-VOICE DATA PACKETS AND AFTER ANY PREVIOUSLY STORED VOICE DATA PACKETS.
504
VOICE DATA PACKET IS COMMUNICATED FROM PACKET BUFFER 307 TO SRAM BUFFER 413 OF VOICE ENGINE PROCESSOR 202 VIA MEDIA INDEPENDENT INTERFACE 203.
505
CPU 401 DEPACKETIZES VOICE DATA PACKET IN SRAM BUFFER 413.
506
SECURITY MODULE 411 DECRYPTS VOICE DATA PACKET, IF NECESSARY.
507
DSP CO-PROCESSOR DECOMPRESSES DEPACKETIZED VOICE DATA.
508
CODEC 414 PERFORMS DIGITAL TO ANALOG CONVERSION ON DECOMPRESSED VOICE DATA.
509
ANALOG VOICE SIGNAL IS COMMUNICATED FROM VOICE ENGINE PROCESSOR 202 TO SPEAKER 212 VIA ANALOG OUTPUT CIRCUIT 431.

FIG. 6

601 ANALOG VOICE SIGNAL IS COMMUNICATED FROM MICROPHONE 211 TO VOICE ENGINE PROCESSOR 202 VIA ANALOG INPUT CIRCUIT 430.

602 CODEC 414 PERFORMS ANALOG TO DIGITAL CONVERSION ON ANALOG VOICE SIGNAL AND STORES DIGITAL VOICE SIGNAL IN SRAM BUFFER 413.

603 DSP CO-PROCESSOR COMPRESSES THE DIGITIZED VOICE SIGNAL.

604 SECURITY MODULE 411 ENCRYPTS THE COMPRESSED VOICE SIGNAL, IF DESIRED.

605 CPU 401 PACKETIZES VOICE SIGNAL.

606 VOICE DATA PACKET IS COMMUNICATED FROM SRAM BUFFER 413 TO PACKET BUFFER 307 OF IP SWITCH CONTROLLER 201 VIA MEDIA INDEPENDENT INTERFACE.

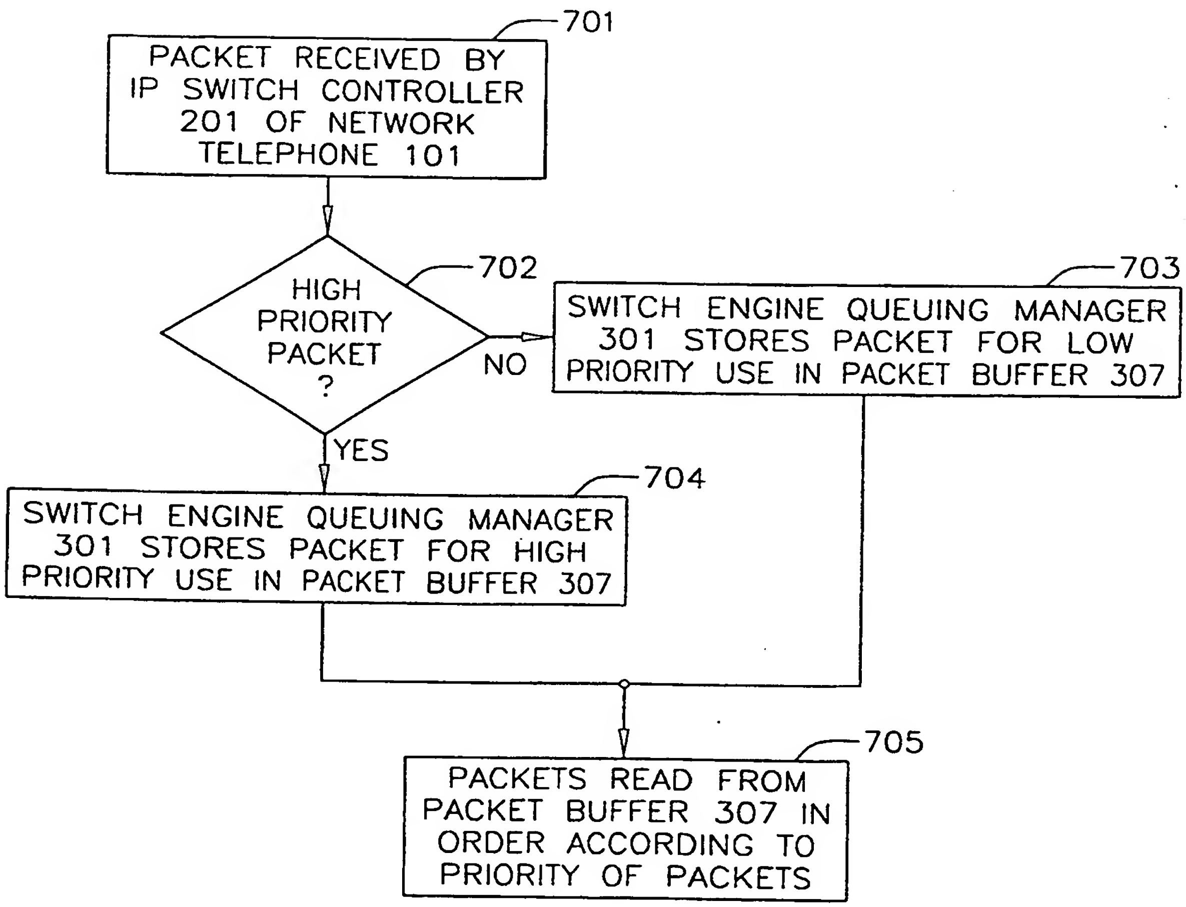
606a Address lookup engine 302 uses DA & VLAN tag, as well as ingress port ID to determine which port(s) to queue frame to.

607 PRIORITY TAG IS APPLIED TO VOICE DATA PACKET BY SWITCH ENGINE QUEUING MANAGER 301 OF IP SWITCH CONTROLLER 201.

608 VOICE DATA PACKET IS STORED BY SWITCH ENGINE QUEUING MANAGER 301 IN PACKET BUFFER 307 FOR READOUT FROM PACKET BUFFER 307 AHEAD OF NON-VOICE DATA PACKETS AND AFTER ANY PREVIOUSLY STORED VOICE DATA PACKETS.

609 VOICE DATA PACKETS ARE TRANSMITTED FROM THE NETWORK TELEPHONE TO THE NETWORK.

FIG. 7


701
PACKET RECEIVED BY IP SWITCH CONTROLLER 201 OF NETWORK TELEPHONE 101
702
HIGH PRIORITY PACKET ?
NO
703
SWITCH ENGINE QUEUING MANAGER 301 STORES PACKET FOR LOW PRIORITY USE IN PACKET BUFFER 307
YES
704
SWITCH ENGINE QUEUING MANAGER 301 STORES PACKET FOR HIGH PRIORITY USE IN PACKET BUFFER 307
705
PACKETS READ FROM PACKET BUFFER 307 IN ORDER ACCORDING TO PRIORITY OF PACKETS

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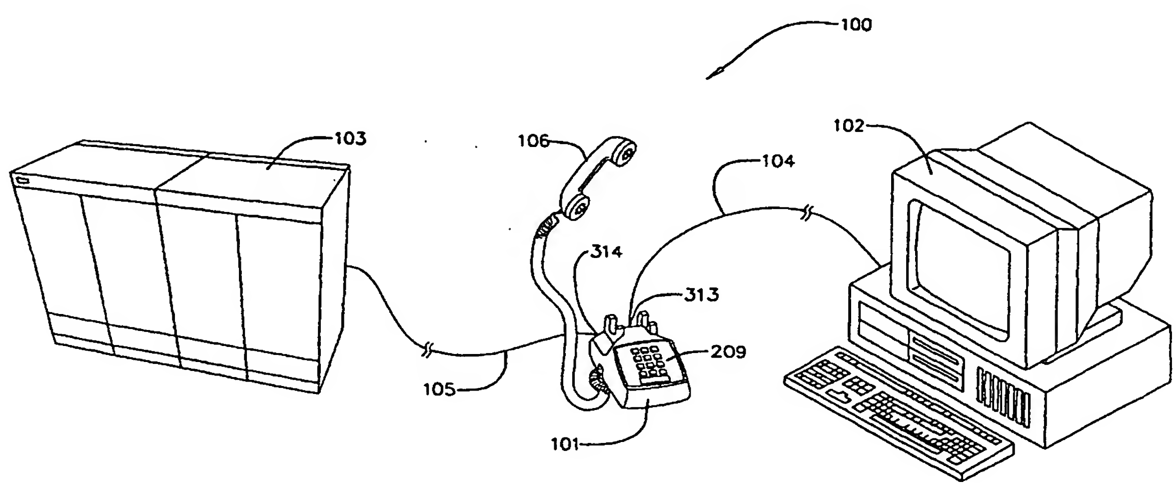

(57) Abstract

The present invention includes a network telephone having a microphone coupled to provide voice data to a network, a speaker coupled to facilitate listening to voice data from the network, a dialing device coupled to facilitate routing of voice data upon the network, a first port configured to facilitate communication with a first network device, a second port configured to facilitate communication with a second network device and a prioritization circuit coupled to apply prioritization to voice data provided by the microphone.

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INTERNATIONAL SEARCH REPORT

International Application No
PC[illegible] 99/28392

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04L12/64

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X Y	EP 0 669 749 A (MULTI TECH SYSTEMS INC) 30 August 1995 (1995-08-30) page 2, line 42 - line 55 page 3, line 42 - line 56 page 7, line 17 - line 19; figure 3 page 7, line 54 -page 8, line 34 page 27, line 12 - line 31 ---	1-4, 69-72 5-11, 73-79
A	EP 0 851 653 A (LUCENT TECHNOLOGIES INC) 1 July 1998 (1998-07-01) column 2, line 40 - line 43 --- -/--	3,4,71, 72

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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Date of the actual completion of the international search	Date of mailing of the international search report
22 May 2000	20. 06. 2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Gregori, S

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International Application No
PCT/US 99/28392

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 4 872 160 A (ULRICH WERNER ET AL) 3 October 1989 (1989-10-03) column 26, line 38 - line 56	35,36, 68,103, 104,136
Y	column 27, line 33 - line 35; figure 20	5-11, 13-19, 21-27, 30,31, 34, 73-79, 81-87, 98,99, 102
X	WO 98 44703 A (ERICSSON TELEFON AB L M) 8 October 1998 (1998-10-08)	12,20, 28,29, 32,33, 80,88, 97,100, 101
Y	page 15, line 20 -page 16, line 7; figure 1	13-19, 21-27, 30,31, 34, 81-87, 98,99, 102
X	WO 98 06201 A (INTELLIGENCE AT LARGE) 12 February 1998 (1998-02-12)	55,123
A	page 14, line 5 -page 15, line 18; figure 4	56-67, 124-135

Box I Observations where certain claims were found unsearchable (Continuation of item 1 of first sheet)

This International Search Report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. [ ] Claims Nos.:
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3. [ ] Claims Nos.:
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This International Searching Authority found multiple inventions in this international application, as follows:

see additional sheet

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FURTHER INFORMATION CONTINUED FROM PCT/ISA/ 210

This International Searching Authority found multiple (groups of) inventions in this international application, as follows:

1. Claims: 1-4,69-72

   Network telephone and method for communicating voice comprising a prioritization circuit

2. Claims: 1,5-9,12-34,36-54,68,69,73-77,80-102,104-122,136

   Network telephone connected to a network switch and method to connect a network telephone to a network switch

3. Claims: 1,10,11,35,69,78,79,103

   Network telephone and method comprising tagging of voice packets

4. Claims: 55-67, 123-135

   Voice engine processor and method comprising a codec, CPU and DSP coprocessor

INTERNATIONAL SEARCH REPORT

Inform[illegible]on patent family members

Inte[illegible]ional Application No
PC[illegible] 99/28392

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